

**MultiVoice features in MAX TNT TAOS 8.0-103**  
*MultiVoice operations*

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### RT-24 (proprietary) codec support

The RT-24 codec is a Lucent Technologies proprietary audio codec that compresses speech samples from 64Kbps pulse code modulation (PCM) to 2.4Kbps, reducing the effective bandwidth required for transmission across the IP network.

This codec uses a 22.5-millisecond audio frame, and encapsulates audio at 8 bytes per frame. The decoder produces 180 samples of audio from the 8-byte encoder output. The RT-24 codec is available for both H.323 VoIP calls and SS7 VoIP calls.

When the RT-24 codec is selected, the MultiVoice Gateway attempts to determine if that codec is supported by the other Gateway during H.245 capability negotiation. If both sides agree to use RT-24 as the preferred codec, both Gateways enable RT-24 on the allocated DSPs to compress and decompress audio after the H.245 open logical channel message is exchanged.

**Note:** RT-24 is a Lucent Technologies proprietary codec, which is available only on MultiVoice Gateways running MAX TNT TAOS 8.0-103. MultiVoice cannot use this codec when communicating with a third-party VoIP gateway.

To enable RT-24 audio processing, set the packet-audio-mode parameter in the default VoIP profile to the selected codec as illustrated by the following:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set packet-audio-mode = rt24
admin> write
VOIP/{ 0 0 } written
```

### G.728 codec support

G.728 is a Low-Delay Code Excited Linear Prediction (LD-CELP) based audio codec that provides toll-quality audio at a bit-rate of 16Kbps. With a frame size of only 2.5 milliseconds, G.728 also has a very low delay. Although the MultiVoice implementation of G.728 uses a frame size of 5 milliseconds, the bitstream from the audio codec is the same as described in the ITU-T standard and can thus be decoded by any G.728 decoder.

Each MultiDSP card supports a maximum of 48 simultaneous G.728 calls for both H.323 VoIP and SS7 VoIP call processing.

When the G.728 codec is selected, the MultiVoice Gateway attempts to determine if the G.728 codec is supported by the other Gateway during H.245 capability negotiation. If both sides agree to use G.728 as the preferred codec, both Gateways use G.728 to compress and decompress audio after the H.245 open logical channel message is exchanged.

**Note:** Although MultiVoice uses a 5-millisecond frame for G.728 processing, it is compatible with any third-party G.728 decoder. However, if a MultiVoice Gateway attempts to communicate with a third-party VoIP gateway transmitting an odd number of 2.5 millisecond frames per IP packet, the call will fail.

When you enable G.728 audio processing in this release the Silence-Det-Cng parameter must be set to no (its default value). The following commands enable G.728 processing:

```
admin> read voip { 0 0 }
VOIP/{ 0 0 } read
admin> set packet-audio-mode = g728
```

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```
admin> set silence-det-cng = no
admin> write
VOIP/{ 0 0 } written
```

## SNMP: Support for the VoIP MIB (ascend 28)

The VoIP MIB enables network management stations to monitor MultiVoice Gateway operations using SNMP. Attributes in the MIB can be obtained by SNMP Get and Get-Next operations. The MIB uses the following object identifiers for identifying MultiVoice Gateway or Gatekeepers to a network manager:

- voipCfgGroup (voipGroup 1)
- voipCfgGkGroup (voipCfgGroup 1)
- voipCfgGwGroup (voipCfgGroup 2)

The MIB uses the following tables for identifying MultiVoice Gatekeeper and Gateway functions.

```
voipCfgGkTable OBJECT-TYPE (voipCfgGkGroup 1)
  SYNTAX SEQUENCE OF VoipCfgGkEntry
  ACCESS not-accessible
  STATUS mandatory
  DESCRIPTION A list of entries for H323 network Gatekeeper.

voipCfgGkEntry OBJECT-TYPE (voipCfgGkTable 1)
  SYNTAX VoipCfgGkEntry
  ACCESS not-accessible
  STATUS mandatory
  DESCRIPTION An entry holding information about the Gatekeeper for
  the system.
  INDEX (voipCfgGkIndex)

VoipCfgGkEntry:
  SEQUENCE :
    voipCfgGkIndex-INTEGER
    voipCfgGkStatus-INTEGER
    voipCfgGkIpAddress-IpAddress

voipCfgGkIndex OBJECT-TYPE ( voipCfgGkEntry 1)
  SYNTAX INTEGER
  ACCESS read-only
  STATUS mandatory
  DESCRIPTION This number uniquely identifies the Gatekeeper.

voipCfgGkStatus OBJECT-TYPE (voipCfgGkEntry 2)
  SYNTAX INTEGER:
    registered(1)
    not_registered(2)
  ACCESS read-only
  STATUS mandatory
  DESCRIPTION This value indicates whether the gateway is registered
  with the Gatekeeper.

voipCfgGkIpAddress OBJECT-TYPE (voipCfgGkEntry 3)
  SYNTAX IpAddress
  ACCESS read-only
  STATUS mandatory
  DESCRIPTION The IP address of the Gatekeeper.
```

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```

voipCfgGwVpnMode OBJECT-TYPE (voipCfgGwGroup 1)
    SYNTAX INTEGER:
        no (1)
        yes(2)
    ACCESS read-only
    STATUS mandatory
    DESCRIPTION Virtual Private Network Toggle Switch.

voipCfgGwPktAudioMode OBJECT-TYPE (voipCfgGwGroup 2)
    SYNTAX INTEGER:
        other(1)
        g711_ulaw(2)
        g711_alaw(3)
        g723(4)
        g729(5)
        g723_6_4kps(6)
    ACCESS read-only
    STATUS mandatory
    DESCRIPTION Audio Coder to be used for voice packetization.

```

The `voipCfgGwVpnMode` and `voipCfgGwPktAudioMode` objects can be accessed using index 0 because they are separate leaves in the MIB tree.

The `voipCfgGkIndex`, `voipCfgGkCurrent` and `voipCfgGkIpAddress` objects are located in the `voipCfgGkTable` table. They can be obtained using `voipCfgGkIndex` as an index.

## SNMP: Traps for VoIP-related conditions

With MAX TNT TAOS 8.0.1, VoIP-enabled MAX TNT units can generate traps for the following MultiVoice Gateway events:

- Change in the call logging server
- Change in configured Gatekeeper for VoIP
- Change in state of a WAN line

For the traps to be sent, traps must be enabled in the system and the individual trap conditions must be set to `yes`. For details about enabling traps, see the *MAX TNT Administration Guide*. Following are the relevant parameters (shown with default values) for enabling the individual trap conditions:

```

[in TRAP/"]
call-log-serv-change-enabled = no
voip-gk-change-enabled = no
wan-line-state-change-enabled = no

```

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<b>Parameter</b>	<b>Specifies</b>
Call-Log-Serv-Change-Enabled	Enable/disable trap generation when the call-logging server changes. If the call-logging server index is changed or if the IP address of the active call-logging server is changed, this trap sends the following information to the SNMP manager: <ul style="list-style-type: none"> <li>• The new call logging server index (callLoggingServerIndex)</li> <li>• The IP address of new call logging server (callLoggingServerIPAddress)</li> <li>• The absolute time to show when the server change occurred (sysAbsoluteCurrentTime) (Ascend Trap 38)</li> </ul>
Voip-GK-Change-Enabled	Enable/disable trap generation when the registered Gatekeeper changes. If a new Gatekeeper is registered with the Gateway, a register request (RRQ) message is sent from the Gateway to the new Gatekeeper. When the Gateway receives the admission request (ARQ) message from the new Gatekeeper, this trap sends the following information to the SNMP manager: <ul style="list-style-type: none"> <li>• The new Gatekeeper index (voipCfgGkIndex)</li> <li>• The IP address of new Gatekeeper (voipCfgGkIpAddress)</li> <li>• The absolute time to show when the Gatekeeper change occurred (sysAbsoluteCurrentTime) (Ascend Trap 39)</li> </ul>
WAN-Line-State-Change-Enabled	Enable/disable trap generation if the state of an E1 or T1 line changes. This trap sends the following information to the SNMP manager: <ul style="list-style-type: none"> <li>• The T1 or E1 line interface index (wanLineIfInde )</li> <li>• The line usage (wanLineUsage). This usage is reported as <b>trunk</b>, <b>quiesced</b>, or <b>disabled</b>.</li> <li>• The absolute time to show when the line state changed (sysAbsoluteCurrentTime) (Ascend Trap 40)</li> </ul>

### **NavisAccess support for VoIP call reporting**

MAX TNT TAOS 8.0.1 supports basic VoIP call reporting using NavisAccess. This includes the capability to generate Start records, Stop records, and Call Progress records for both VoIP and fax calls. These records allow NavisAccess to monitor Gateway resource usage and provide information to create billing records. Each VoIP call can generate two or more records.

#### *Start records*

A Start record reports the point in the call where a speech communications is established. Start records can provide the following information:

<b>Attribute</b>	<b>Specifies</b>
Ascend-Call-Direction	Direction of the call between the Gateway and PSTN. The reported values are Ascend-Call-Direction-Incoming (0) and Ascend-Call-Direction-Outgoing (1). (Ascend Trap 48)
NAS-Port	Encoded NAS port used for this call. (RFC Trap 5)

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<b>Attribute</b>	<b>Specifies</b>
NAS-Port-Type	Encoded NAS port used for this call. The value 7 for this attribute identifies a VoIP call. (RFC Trap 61)
NAS-IP-Address	NAS IP address associated with this call. (RFC Trap 4)
Session-Id	NAS session index recorded in the session table for this call. (RFC Trap 44)
Ascend-Modem-PortNo	DSP/modem port allocated for processing this call. This value is part of the resource count information, and is repeated each time it is allocated for a call. (Ascend Trap 120)
Ascend-Modem-SlotNo	Slot where the DSP/modem card associated with the reported Ascend-Modem-PortNo is located. This value is part of the resource count information, and is repeated each time it is allocated for a call. (Ascend Trap 121)
Ascend-Modem-ShelfNo	Shelf where DSP/modem card allocated for processing this call is installed. This is part of the resource count information, and is repeated each time it is allocated for a call. (Ascend Trap 122)
Called-Station-Id (DNIS)	Dialed number string reported by the Gateway for the called destination. (RFC Trap 30)
Ascend-Dialed-Number	Dialed number string used by the Gateway to complete the call. (Ascend Trap 24)
Service-Type	Requested type of service, the value of the Type of Service byte, for this call. (RFC Trap 6)
Ascend-H323-Destination-NAS-ID	NAS IP address used to route the call to the connecting Gateway. (Ascend Trap 22)
Ascend-H323-Gatekeeper-IP	IP address of the Gatekeeper used to route the call. The Gateway is registered with this Gatekeeper. (Ascend Trap 19)
Ascend-Global-Call-Id	IP address used by the Gatekeeper to identify the connecting Gateway for this call. (Ascend Trap 20)
Ascend-H323-Conference-ID	IP address used to identify the called destination. (Ascend Trap 21)
Ascend-H323-Presession-Time	Time from the moment the caller finishes dialing the destination telephone number until the moment the speech path is established to the called destination. (Ascend Trap 198)
Ascend-H323-Dialed-Time	Time the user spends dialing the destination telephone number. This value will be zero for call originating from the LAN. (Ascend Trap 23)
Ascend-Session-Type	Audio codec used for processing the call. (Ascend Trap 18)

**Stop records**

A Stop record is generated at the moment when MultiVoice begins to tear down the speech path or when an incoming call to a Gateway fails to connect. A Start record can contain following information:

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<b>Attribute</b>	<b>Specifies</b>
Acct-Session-Time	Time from the moment the speech path is established to the called destination until the moment MultiVoice begins to tear down the speech path. (RFC Trap 46)
Ascend-Connect-Progress	A number that represents the call connect state at the time the call was terminated. (Ascend Trap 195)
Ascend-Disconnect-Cause	A number that reports the H.323 call disconnection reason. (Ascend Trap 196)
Ascend-H323-Inter-Arrival-Jitter	Estimated interarrival jitter for voice packets received by a Gateway. (Ascend Trap 25)
Ascend-Dropped-Octets	The number of voice frames (in bytes) dropped by a Gateway during call processing. (Ascend Trap 26)
Ascend-Dropped-Packets	Number of voice packets dropped by a Gateway during call processing. (Ascend Trap 26)
Acct-Input-Octets	Number of voice frames (in bytes) received by a Gateway during this call. (RFC Trap 42)
Acct-Input-Packets	Number of voice packets received by a Gateway during this call. (RFC Trap 47)
Acct-Output-Octets	Number of voice frames (in bytes) sent by a Gateway during this call. (RFC Trap 43)
Acct-Output-Packets	Number of voice packets sent by a Gateway during this call. (RFC Trap 48)

*Call Progress records*

A Call Progress record can be generated during a VoIP call when a change in resource occurs for a fax or transparent modem call. For fax calls, this record includes the modem speed and modulation. A progress message contains all the information included in a Start record.

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# **EXHIBIT 8**

**Lucent Technologies**  
Bell Labs Innovations



# **MultiVoice® for APX™/MAX TNT®**

Configuration Guide

Part Number: 7820-0651-005  
For software version 10.0  
July 2002

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# About This Guide

This guide provides instructions on how to configure an APX™ or MAX TNT® to process MultiVoice voice over IP (VoIP) calls.



**Note** This manual describes the full set of features for units running software version TAOS 10.0. Some features might not be available with earlier versions or specialty loads of the software.

The APX family of products includes multiple platforms that differ in call capacity and hardware, but support the same operating system and similar configuration options. The APX family, which includes the APX 8000 and APX 1000 products, shares many features with its MAX TNT predecessor. For features that are supported with no differences across all the platforms, this manual often refers to your product as a *TAOS unit*.



**Warning** Before installing your unit, be sure to read the safety instructions in the *Edge Access and Broadband Access Safety and Compliance Guide*. For information specific to your unit, see the "Safety-Related Electrical, Physical, and Environmental Information" appendix in your unit's hardware installation guide or *Getting Started Guide*.

## What you need to know

This manual is intended for the person who configures and maintains your TAOS unit running MultiVoice. To use the manual effectively, you must have a basic understanding of security and configuration, and be familiar with authentication servers and networking concepts. You also need to understand Internet and telecommuting concepts and dial-in connections (both framed protocol sessions and user logins).

Following are the special characters and typographical conventions used in this manual:

Convention	Meaning
Monospace text	Represents text that appears on your computer's screen, or that might appear on your computer's screen.
Boldface monospace text	Represents characters that you enter exactly as shown (unless the characters are also in <i>italics</i> —see <i>Italics</i> , following). If you can enter the characters but are not specifically instructed to, they do not appear in boldface.

**About This Guide**

<b>Convention</b>	<b>Meaning</b>
<i>Italics</i>	Represent variable information. Do not enter the words themselves in the command. Enter the information they represent. In ordinary text, italics are used for titles of publications, for some terms that would otherwise be in quotation marks, and to show emphasis.
[ ]	Indicate an optional argument you might add to a command. To include such an argument, type only the information inside the brackets. Do not type the brackets unless they appear in boldface.
	Separates command choices that are mutually exclusive.
>	Separates levels of profiles, subprofiles, and parameters in a hierarchical menu when the path to a menu item is referred to in text.
:	Separates levels of profiles, subprofiles, and parameters in a pathname displayed in the command-line interface or referred to in text.
Key1+Key2	Represents a combination keystroke. To enter a combination keystroke, press the first key and hold it down while you press one or more other keys. Release all the keys at the same time. (For example, Ctrl+H means hold down the Ctrl key and press the H key.)
Press Enter	Means press the Enter or Return key or its equivalent on your computer.
	Introduces important additional information.
<b>Note:</b>	
	Warns that a failure to follow the recommended procedure can result in loss of data or damage to equipment.
<b>Caution:</b>	
	Warns that a failure to take appropriate safety precautions can result in physical injury.
<b>Warning:</b>	
	Warns of danger of electric shock.
<b>Warning:</b>	

**Documentation set**

The documentation set for APX and MAX TNT products consists of the following manuals, available at <http://www.lucent.com/support> and <http://www.lucentdocs.com/ins>:

- **Read me first:**

[About This Guide](#)

- *Edge Access and Broadband Access Safety and Compliance Guide*. Contains important safety instructions and country-specific compliance information that you must read before installing a unit.
- *TAOS Command-Line Interface Guide*. Introduces the TAOS command-line environment and shows how to use the command-line interface effectively. This manual describes keyboard shortcuts and introduces commands, security levels, profile structure, and parameter types.
- **Installation and basic configuration:** *Getting Started Guide* or hardware installation guide for your unit. Shows how to install the unit's chassis and hardware, and includes technical specifications. A *Getting Started Guide* also shows you how to provide the basic configuration needed to access the unit on a network.
- **Configuration:**
  - *Physical Interface Configuration Guide* for your unit. Describes how to provision the slot cards supported in the unit, and how to configure the cards' physical interfaces. This guide also describes system allocation of slot card resources, and how to use the supported cards in a variety of data environments.
  - *APX/MAX TNT ATM Configuration Guide*. Describes how to configure Asynchronous Transfer Mode (ATM) permanent virtual circuit (PVC) and switched virtual circuit (SVC) ATM interfaces. It includes information about ATM direct and ATM-frame relay circuits.
  - *APX/MAX TNT Frame Relay Configuration Guide*. Describes how to configure frame relay operations on a unit. This guide explains physical layer restrictions and how to create permanent virtual circuit (PVC) and switched virtual circuit (SVC) interfaces. It includes information about Multilink frame relay (MFR) and link management, as well as frame relay and frame relay direct circuits.
  - *APX/MAX TNT WAN, Routing, and Tunneling Configuration Guide*. Shows how to configure LAN and WAN routing for analog and digital dial-in connections on a unit. This guide includes information about IP routing, Open Shortest Path First (OSPF) routing, Border Gateway Protocol (BGP) routing, Internet Group Management Protocol (IGMP) routing, multiprotocol routers, virtual routers (VRouters), and tunneling protocols.
- **MultiVoice:**
  - *MultiVoice® for APX/MAX TNT Configuration Guide*. Shows how to configure the MultiVoice® application to run on a unit in both Signaling System 7 (SS7) and H.323 Voice over IP (VoIP) configurations.
  - *MultiVoice Access Manager User's Guide*. Describes the installation, configuration, and administration of MultiVoice Access Manager, which provides H.323 gatekeeper functions for MultiVoice networks.
- **RADIUS:** *TAOS RADIUS Guide and Reference*. Describes how to set up a unit to use the Remote Authentication Dial-In User Service (RADIUS) server, and contains a complete reference to RADIUS attributes.
- **Administration and troubleshooting:** *APX/MAX TNT Administration Guide*. Describes how to administer a unit, including how to monitor the system and cards, troubleshoot the unit, and configure the unit to use the Simple Network Management Protocol (SNMP).

**About This Guide**


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- **Reference:**
  - *APX/MAX TNT Reference*. An alphabetic reference to all commands, profiles, and parameters supported on a unit.
  - *TAOS Glossary*. Defines terms used in the documentation for a unit.

**Related publications**

This guide and documentation set do not provide a detailed explanation of products, architectures, or standards developed by other companies or organizations. Following are some publications that you might find useful:

- ITU Telecommunication sector standard (ITU-T) H.323, *Packet-based multimedia communications systems* (Feb. 1998), International Telecommunications Union.
- RFC 1889, *RTP: A Transport Protocol for Real-Time Applications* (Jan. 1996), IETF.
- RFC 2705, *Media Gateway Control Protocol (MGCP)* (Oct. 1999), IETF.
- *Signaling in Today's Telecommunication Networks*, John G. van Bosse.
- *Delivering Voice over IP Networks*, Dan Minoli, Emma Minoli, Daniel Minoli.
- *Delivering Voice Over Frame Relay and ATM*, Dan Minoli.
- *The Guide to T1 Networking*, William A. Flanagan.
- *TCP/IP Illustrated*, W. Richard Stevens.
- *Firewalls and Internet Security*, William R. Cheswick and Steven M. Bellovin.

Following are some related World Wide Web (WWW) sites:

- <http://www.ietf.org/rfc>
- <http://www.itu.ch/>
- <http://www.cs.columbia.edu/~hgs/rtp/drafts/VoIP97-8.pdf>
- <http://www.cs.columbia.edu/~hgs/rtp/>

**Note** The listed web sites were available at the time of this manual's publication. Lucent does not maintain these sites and cannot guarantee their availability in the future.



# Introducing MultiVoice Concepts

1

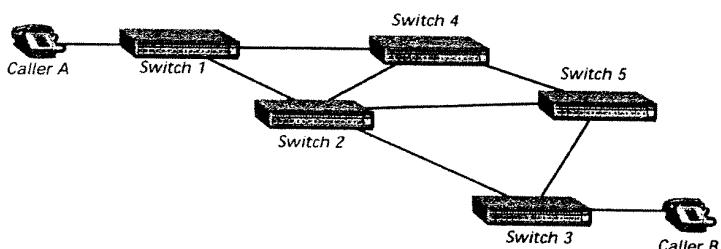
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## The public switched telephone network

Traditionally, real-time voice information is sent over the public switched telephone network (PSTN). Circuit-switched technology provides every call with dedicated bandwidth, usually 64Kbps. End-to-end calls are established on the basis of a sequence of dialed digits, and the PSTN dedicates a physical path between callers. Because the telephone equipment establishes the call path at the beginning of the call, the path can change *between* calls, but never while a call is active.

Figure 1-1 illustrates an example of a PSTN network. Caller A dials Caller B's phone number. As Caller A dials the phone number, the network might route the call from Switch 1 to Switch 2 to Switch 3, which connects to Caller B. Once the PSTN establishes the call, communication travels only through Switch 1, Switch 2, and Switch 3.

Figure 1-1. Example of call routing over circuit-switched PSTN



If Caller A dials Caller B again, the PSTN might establish the call by routing it from Switch 1 to Switch 4 to Switch 5 to Switch 3 before finally connecting Caller A to Caller B. Again, the path can change between calls, but not during any specific call.

In contrast, an Internet Protocol (IP) network has a packet-switched architecture. Devices transmit data in packets, and the path of one packet from end to end can vary from another packet within an established session. In addition to data, packets

**Introducing MultiVoice Concepts***The MultiVoice network*

contain addressing information, which routing devices use to send each packet to its destination. Routing devices maintain tables that instruct them how to direct packets. Dynamic protocols, like Routing Information Protocol (RIP) or Open Shortest Path First (OSPF), define methods that routing devices use to update each other as networking environments change.

In the past, the PSTN was the only network supporting voice communication. With MultiVoice, voice traffic can be transmitted across IP networks.

## The MultiVoice network

MultiVoice complies with International Telecommunications Union Telecommunication Standardization sector standard (ITU-T) H.323 for transmitting voice telephone calls across IP networks. The H.323 standard defines a framework for the transmission of real-time voice communications across IP networks. MultiVoice on the APX or MAX TNT also supports integration with Signaling System 7 (SS7) networks by means of IP Device Control (IPDC), a media gateway control protocol, to provide call control for Voice over IP calls originating from SS7 networks.

### Multivoice terms and definitions

In addition to the vocabulary used in the TAOS environment, MultiVoice uses some specific voice-related expressions. The following lists the most common terms with their definitions.

Term	Definition
Call end-point	The communications device used to initiate or answer a call, or a call's origin or destination.
Egress	A general voice-related term for an exit. For MultiVoice, a location or device used to route data from the packet network onto the analog network.
Egress gateway	Related terms: egress PSTN, egress switch A term specific to MultiVoice for the TAOS unit which connects a VoIP call to the called telephone number. The egress gateway:
	<ul style="list-style-type: none"> <li>• Dials the destination telephone number</li> <li>• Converts data from the packet network to analog voice</li> <li>• Reports call progress</li> </ul>
	Related terms: egress MultiVoice Gateway, egress TAOS unit
Far end	A general voice-related term for the remote call termination point, relative to the active call end point. For MultiVoice, the location or device—relative to the point-of-origin of network packets, call signals, etc.—where packets, call signals, etc., for the remote call end point are processed.
	Related terms: far-end PSTN, far-end switch
Far-end gateway	(Specific to MultiVoice) The TAOS unit at the opposite end of the packet network connection—relative to the active call end point.
	Related terms: far-end MultiVoice Gateway, far-end TAOS unit

**Introducing MultiVoice Concepts**  
*The MultiVoice network*

<b>Term</b>	<b>Definition</b>
Ingress	(General) An entrance. For MultiVoice, a location or device where voice signals from the analog network are routed onto the packet network.  Related terms: ingress PSTN, ingress switch
Ingress gateway	(Specific to MultiVoice) The TAOS unit where a VoIP call originates. The ingress gateway: <ul style="list-style-type: none"> <li>• Accepts calls from the PSTN</li> <li>• Initiates requests for call admissions</li> <li>• Converts analog voice to packet network data</li> <li>• Reports call progress</li> </ul> Related terms: ingress MultiVoice Gateway, ingress TAOS unit
MultiVoice Access Manager (MVAM)	A MultiVoice component that supports the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) H.323 standard for managing an IP network. MVAM supports MultiVoice Gateways, user profiles, and authentication.  Capabilities supported by MVAM include phone-to-IP address translation, Web-based administration interface, PIN-based user authentication, virtual private network (VPN) support, Telephone number aliases, call detail reporting (CDR), Gateway and user database support, and third-party billing system support.
MultiVoice Gateway	A MultiVoice component that supports the International Telecommunication Union Telecommunication Standardization Sector (ITU-T) H.323 standard for transmitting voice over an IP network.  When a voice call is received at a local MultiVoice Gateway, the voice signal is packetized, compressed, and transmitted over the packet network using standard protocols and voice-compression technologies.  At the remote gateway, the process is reversed and the call is delivered over the remote packet network to its intended destination.
Near end	(General) A local call termination point, relative to the active call end point. For MultiVoice, the location or device—relative to the point-of-origin of network packets, call signals, etc.—where packet processing, call signaling, etc., is initiated for the active call end point.  Related terms: near-end PSTN, near-end switch
Near-end gateway	(Specific to MultiVoice) The TAOS unit which provides the local connection to the packet network—relative to the active call end point.  Related terms: near-end MultiVoice Gateway, near-end TAOS unit